

ABSTRACT OF THE DISCLOSURE

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A speech coding apparatus includes a spectrum parameter calculation section, an adaptive codebook section, a sound source quantization section, a discrimination section, and a multiplexer section. The spectrum parameter calculation section receives a speech signal and quantizes a spectrum parameter. The adaptive codebook section obtains a delay and a gain from a past quantized sound source signal using an adaptive codebook, and obtains a residue by predicting a speech signal. The sound source quantization section quantizes a sound source signal using the spectrum parameter. The discrimination section discriminates the mode. The sound source quantization section has a codebook for representing a sound source signal by a combination of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses in a predetermined mode, and searches combinations of code vectors and shift amounts used to shift the positions of the pulses to output a combination of a code vector and shift amount which minimizes distortion relative to input speech. The multiplexer section outputs a combination of outputs from the spectrum parameter calculation section, the adaptive codebook section, and the sound source quantization section.